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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/695,246	10/27/2003	Gillray L. Kandel	0200.002.01/US	4321

7590 12/20/2007

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EXAMINER	
KOVACEK, DAVID M	
ART UNIT	PAPER NUMBER
2626	
MAIL DATE	DELIVERY MODE
12/20/2007	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)
	10/695,246	KANDEL ET AL.
	Examiner	Art Unit
	David Kovacek	2626

-- *The MAILING DATE of this communication appears on the cover sheet with the correspondence address* --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 27 October 2003.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-14 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-14 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on 27 October 2007 is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date 10/27/2003.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Claim Objections

1. **Claim 8** is objected to because of the following informalities:

- The sixth limitation of **claim 8** contains incorrect grammar and should be corrected. The examiner suggests instead that this limitation read, "[the] controlling the gain for the amplification [to be] based on the level sensed..."
- Appropriate correction is required.

Claim Rejections - 35 USC § 112

The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

2. **Claim 8** is rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

Regarding **claim 8**, the second limitation refers to, "differentially amplifying a first frequency range...and a second frequency range." This is ambiguous because it is unclear whether it is directed to the use of differential amplifiers for amplifying the first and/or second frequency range, or rather if the two frequency ranges are amplified using different means or techniques. In light of

the specification, the latter usage seems more appropriate. The examiner has selected this interpretation for the purposes of examination and application of prior art.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. **Claims 1**, **6**, **8**, and **10-13** are rejected under 35 U.S.C. 103(a) as being unpatentable over US Patent 5,506,910, listed in IDS, hereinafter referred to as Miller.

Regarding **claim 1**, Miller discloses an amplifying system, comprising:

- a first amplifying circuit to linearly amplify a first frequency range of an audio signal that substantially comprises first speech formant frequencies (Fig. 1, elements 20, 32; Fig. 3, element 30; Col. 4, lines 211-30),
- a second amplifying circuit to linearly amplify a second frequency range of the audio signal that substantially comprises second speech formant

frequencies (Fig. 1, elements 20, 32; Fig. 3, element 30; Col. 4, lines 211-30);

It is noted by the examiner that though Miller does not explicitly specify two individual amplifying circuits, this is obviously analogous to Miller's disclosure of a multiband gain control (Fig. 1, element 32; Col. 4, lines 21-24), including the potential of analog implementation of the multiband gain control. One skilled in the art at the time the invention was made would recognize such a device as a combination of amplifying circuits for amplifying different ranges of audio frequencies differently as is claimed.

- the amplification of the first frequency range and the amplification of the second frequency range to emulate at least one acoustic property of a passive device (Fig. 2; Col. 3, line 64 - Col. 4, line 10);

It is noted by the examiner that the broadest reasonable interpretation of a "passive device" according to one of ordinary skill in the art at the time of the invention would include an acoustic space such as the room in which a system operates. Miller implies the emulation of a room with a perfectly neutral frequency response (Col. 4, lines 5-6), as well as further disclosing the ability to replicate any frequency response the user desires.

- a mixer [mixer/preamplifer] to combine the first frequency range and the second frequency range into an amplified audio signal (Fig. 1, element 24; Col. 3, lines 38-39); and

- an acoustic output device [speaker system] to transmit [broadcast] the amplified audio signal [program signal] (Fig. 1, element 36; Col. 3, lines 44-45).

Regarding **claim 6,** Miller discloses a public announcement system, comprising:

- a first amplifier to linearly amplify a first frequency range of an audio signal, the first frequency range substantially of first speech formant (Fig. 1, elements 20, 32; Fig. 3, element 20; Col. 4, lines 21-30);
- a second amplifier to linearly amplify a second frequency range of the audio signal, the second frequency range substantially of second speech formant (Fig. 1, elements 20, 32; Fig. 3, element 20; Col. 4, lines 21-30);
- the amplification of the first frequency range and the amplification of the second frequency range weighted differently (Fig. 1, element 32; Col. 4, lines 21-30)
- and to emulate at least one acoustic property of a passive device (Fig. 2; Col. 3, line 64 – Col. 4, line 10);
- a mixer [mixer/preamplifier] to combine the signal amplified by the first amplifier of the first frequency range and the signal amplified by the second amplifier of

the second frequency range into an amplified audio signal (Fig. 1, element 36; Col. 4, lines 6-10); and

- an acoustic output device [speaker system] to transmit [broadcast] the amplified audio signal [program signal] (Fig. 1, element 36; Col. 3, lines 44-45).

It is noted by the examiner that this in many respects similar to **claim 1** and is largely rejected for the same reasons. It is further noted by the examiner that changes in the preamble regarding intended use are not given patentable weight because the recitation occurs in the preamble. A preamble is generally not accorded any patentable weight where it merely recites the purpose of a process or the intended use of a structure, and where the body of the claim does not depend on the preamble for completeness but, instead, the process steps or structural limitations are able to stand alone. See *In re Hirao*, 535 F.2d 67, 190 USPQ 15 (CCPA 1976) and *Kropa v. Robie*, 187 F.2d 150, 152, 88 USPQ 478, 481 (CCPA 1951).

. One significant difference between the claims is the slightly narrower scope required by the inclusion of the amplification of the first frequency range and the amplification of the second frequency range weighted differently in **claim 6**. However, it is further noted that Miller discloses this as well in exemplifying a number of uses for the multigain control (Col. 4, lines 06-10), some of which comprise conditions

where different frequency ranges are weighted differently [increased or decreased bass or treble].

Regarding **claim 8**, Miller discloses a method of enhancing speech [audio] in a public address system, comprising:

- receiving an audio signal (Fig. 1, elements 24, 26, 28, 30, 70; Col. 3, lines 38-42; Col. 5, line 67 – Col. 6, line 03);
- differentially amplifying a first frequency range that substantially consists of first speech formant frequencies and a second frequency range that substantially consists of second formant frequencies of the audio signal (Fig. 1, elements 20, 32; Fig. 3, element 20; Col. 4, lines 21-30);

It is noted by the examiner that though Miller does not explicitly disclose the differentiation between frequency ranges as relating directly to speech formant frequencies, the multigrain control unit is directed to the full range of human auditory frequencies, which necessarily includes ranges that comprise the frequency ranges related to the first and second formants of human speech.

- mixing an injected inaudible signal tone [masked sine wave] with the audio signal (Fig. 1, element 22; Col. 2, lines 33-38, lines 46-47; Col. 4, lines 64-67);

- sensing a level of the signal tone within the audio signal received (Fig. 1, element 42; Col. 3, lines 49-52); and
- controlling a gain for amplification of the second frequency range based on the level of the signal tone sensed (Fig. 1, element 44; Col. 3, line 67 – Col. 4, line 06);
- the controlling the gain for the amplification to be based on the level sensed (Col. 3, lines 54-60; Col. 4, lines 21-25),
- to substantially prevent regenerative oscillation [unwanted acoustic feedback/howl] of the audio signal (Fig. 3, element 62; Col. 7, lines 09-16) and
- to amplify the second formant frequencies without creating howling (Col. 3, lines 54-60; Col. 7, lines 54-60).

Regarding **claim 10**, Miller discloses or renders obvious all limitations of **claim 8** as applied above, and further discloses that the sensing of the signal tone makes use of a narrow band filter (Fig. 1, element 42; Col. 5, lines 04-11, lines 30-39).

Regarding **claim 11**, Miller discloses or renders obvious all limitations of **claim 8** as applied above, and further discloses sensing a change in at least one environmental variable (Col. 4, lines 54-60; Col. 4, line 67 - Col. 5, line 06; Col.

5, lines 39-47), and further implies the controlling the gain for the amplification is further based on the sensed change (Col. 3, line 64 – Col. 4, line 10; Col. 4, lines 54-63).

Regarding **claim 12**, Miller discloses or renders obvious all limitations of **claim 11** as applied above, and further implies that the sensed change is based on the signal tone (Col. 3, lines 54-60; Col. 3, line 64 – Col. 4, line 10).

Regarding **claim 13**, Miller discloses or renders obvious all limitations of **claim 8** as applied above, and further discloses suggests that the differentially amplifying emulates at least one acoustic property of a passive device (Fig. 2; Col. 3, lines 61-64; Col. 3, line 64 - Col. 4, line 10).

4. **Claims 2-4**, **7** and **14** are rejected under 35 U.S.C. 103(a) as being unpatentable over Miller in view of *Acoustic Systems in Biology*, hereinafter referred to as Fletcher.

Regarding **claim 2**, Miller discloses all limitations of **claim 1** as applied above, but does not disclose nor render obvious the emulation of either an ear cupping or ear trumpet.

Fletcher provides equations which predict the frequency response of a trumpet horn of known dimensions (p. 192-193), which could be used by one of ordinary skill in the art at the time the invention was made to emulate the frequency response of such a

horn (for example, an ear trumpet) using a multiband gain control method, such as is taught in Miller. The references are combinable because each deals with the analysis and adjustment of audible sound. Fletcher further provides motivation to combine in disclosing the similarities between horn shapes and the pinna in animal ears including humans (Page 178, pages 200-201). Therefore, the examiner contends that it would have been obvious to one of ordinary skill in the art at the time the invention was made to use the teachings of Fletcher to modify the teachings of Miller in order to implement an amplifying system with multiband amplification that emulates an ear trumpet for the purpose of providing a close approximation to a human ear's frequency response in the system.

Regarding **claim 3**, Miller in view of Fletcher discloses all limitations of **claim 2** as applied above, and Miller further discloses:

- a receiver to receive an input signal and to source therefrom the audio signal of the first and second frequency ranges (Fig. 1, element 40; Col. 3, lines 45-49);
- a generator to generate an injection tone (Fig. 1, element 22; Col. 2, lines 33-38, lines 46-47; Col. 4, lines 64-67);
- the mixer to combine the injection tone with the signals of the first and the second frequency ranges

amplified by the respective first and the second amplifiers (Fig. 1, element 22; Col. 3, lines 34-36); and

It is noted by the examiner that Miller does not disclose the injection tone to be combined at the mixer, but instead with a separate masked sine wave adder.

It is further noted that one of ordinary skill in the art would recognize that modifying the masked sine wave adder to act as an additional input to mixer/preamplifier 24 would produce the same results with a reasonable expectation of success. A person of ordinary skill in the art, upon reading the teachings Miller, would also have recognized the motivation to modify exists because the masked sine wave adder 22 would then be accessible to allow manual control from the user to improve performance of tone detection in unexpected conditions. Furthermore, this modification is a known option within the technical grasp of one of ordinary skill in the art at the time the invention was made.

Therefore, the examiner contends that it would have been obvious to one of ordinary skill in the art at the time the invention was made to make this modification to the teachings of Miller in order to allow manual control of the tone injection by the user to improve performance of the tone detection in unexpected conditions with reasonable expectation of success.

- the acoustic output device to transmit the amplified audio signal of the first and the second frequency ranges together with the injection tone (Fig. 1, element 34; Col. 3, lines 42-45); and
- a detector to recover a portion of the injection tone signal feedback and received by the receiver in the input signal (Fig. 1, element 42; Col. 3, lines 49-52; Col. 5, lines 15-19);
- the second amplifier comprising an adjustable gain of a magnitude controlled dependent on the level of the injection tone signal recovered by the detector (Fig. 1, element 44; Col. 3, line 67 – Col. 4, line 06; Col. 5, lines 36-39).

Regarding **claim 4**, Miller in view of Fletcher discloses all limitations of **claim 3** as applied above, and Miller further discloses that the generator [sine wave adder] is intended to inject a tone that is not audible [masked] (Col. 4, line 65 - Col. 5, line 15).

Regarding **claim 7**, this claim is very similar to **claim 2**, in that it includes the same limitation of **claim 2** and only further differs in the preamble from the base **claims 1** and **6** as applied above. For these reasons, **claim 7** is rejected for the same reasons as applied to **claim 2**.

Regarding **claim 14**, Miller discloses or renders obvious all limitations of **claim 8** as applied above, but does not disclose nor render obvious the emulation of either an ear cupping or ear trumpet.

Fletcher provides equations which predict the frequency response of a trumpet horn of known dimensions (p. 192-193), which could be used by one of ordinary skill in the art at the time the invention was made to emulate the frequency response of such a horn (for example, an ear trumpet) using a multiband gain control method, such as is taught in Miller.

Further because **claim 8** claims subject matter very similar to that of **claim 1** and **claim 14** only provides limitations presented in **claim 2**, the motivation to combine the references with regard to **claim 2** is applicable to **claim 14**.

5. **Claim 9** is rejected under 35 U.S.C. 103(a) as being unpatentable over Miller in view of US Patent 4,539,692 hereinafter referred to as Munter.

Regarding **claim 9**, Miller discloses all limitations of **claim 8** as applied above, but does not disclose the modulation of the signal tone using at least one of pulse modulation and frequency modulation.

Munter discloses a system of automatic gain control in a voice transmission circuit, such as a telephone system, that includes a inaudible control signal using pulse modulation (Col. 4, lines 05-12, lines 21-25, lines 31-34).

The two references are combinable because each is directed to a system of transmitted and reproducing audio data with controllable gain. Munter provides motivation in disclosing the utility of identifiable pulse code modulated signals to prompt a third-party network system in order to automate the process for increased speed and accuracy (Col. 1, lines 18-25).

Therefore, the examiner contends that it would have been obvious for one of ordinary skill in the art to modify the teachings of Miller using the teachings of Munter in order to implement an automatic equalization system capable of detecting and identifying pulse code modulated signals in order to automate operation via a third-party network system.

6. **Claim 5** is rejected under 35 U.S.C. 103(a) as being unpatentable over Miller in view of Fletcher and in further view of Munter.

Regarding **claim 5**, Miller in view of Fletcher discloses all limitations of **claim 3** as applied above, but does not adequately teach a predetermined modulation encoding or decoding for the injection tone signal.

Munter discloses a system of automatic gain control in a voice transmission circuit, such as a telephone system, that includes a inaudible control signal using pulse modulation (Col. 4, lines 05-12; lines 21-25, lines 31-34).

The three references are combinable because each is directed to a system of transmitted and reproducing audio data with controllable gain. Fletcher provides

motivation to combine in disclosing the similarities between horn shapes and the pinna in animal ears including humans (Page 178, pages 200-201). Munter further provides motivation in disclosing the utility of identifiable pulse code modulated signals to prompt a third-party network system in order to automate the process for increased speed and accuracy (Col. 1, lines 18-25).

Therefore, the examiner contends that it would have been obvious for one of ordinary skill in the art to modify the teachings of Miller in view of Fletcher using the teachings of Munter in order to implement an automatic equalization system that emulates an ear trumpet for the purpose of providing a close approximation to a human ear's frequency response that is further capable of detecting and identifying pulse code modulated signals in order to automate operation via a third-party network system.

Double Patenting

7. The nonstatutory double patenting rejection is based on a judicially created doctrine grounded in public policy (a policy reflected in the statute) so as to prevent the unjustified or improper timewise extension of the "right to exclude" granted by a patent and to prevent possible harassment by multiple assignees. A nonstatutory obviousness-type double patenting rejection is appropriate where the conflicting claims are not identical, but at least one examined application claim is not patentably distinct from the reference claim(s) because the examined application claim is either anticipated by, or would have been obvious over, the reference claim(s). See, e.g., *In re Berg*, 140 F.3d 1428, 46 USPQ2d 1226 (Fed. Cir. 1998); *In re Goodman*, 11 F.3d 1046, 29 USPQ2d 2010 (Fed. Cir. 1993); *In re Longi*, 759 F.2d 887, 225 USPQ 645 (Fed. Cir. 1985); *In re Van Ornum*, 686 F.2d 937, 214 USPQ 761 (CCPA 1982); *In re Vogel*, 422 F.2d 438, 164 USPQ 619 (CCPA 1970); and *In re Thorington*, 418 F.2d 528, 163 USPQ 644 (CCPA 1969).

A timely filed terminal disclaimer in compliance with 37 CFR 1.321(c) or 1.321(d) may be used to overcome an actual or provisional rejection based on a nonstatutory double patenting ground provided the conflicting application or patent either is shown to

be commonly owned with this application, or claims an invention made as a result of activities undertaken within the scope of a joint research agreement.

Effective January 1, 1994, a registered attorney or agent of record may sign a terminal disclaimer. A terminal disclaimer signed by the assignee must fully comply with 37 CFR 3.73(b).

8. **Claims 8-14** are rejected on the ground of nonstatutory obviousness-type double patenting as being unpatentable over **claims 1-8** of U.S. Patent No. 6,647,123. Although the conflicting claims are not identical, they are not patentably distinct from each other because the subject matter claimed in the instant application is fully disclosed in the patent and is covered by the patent since the patent and the application are claiming common subject matter, as follows:

Claims 8-14 contains subject matter that is an obvious variation of subject matter already existing in **claim 1-8** of US Patent 6,647,123. The table below provides an example of the comparison between the claim language that exists between the two documents for **claims 8** and **1**.

CURRENT APPLICATION	US PATENT 6,647,123	COMPARISON
8. A method of enhancing speech intelligibility in a public address system, comprising:	1. A method of processing an audio signal in a hearing aid for increasing speech intelligibility to a human, comprising:	The preambles of each claim do not carry patentable weight because each only provides intended use without directly affecting the limitations that follow.
receiving an audio signal;	receiving an audio signal;	These limitations are identical.
differentially amplifying a first frequency range that substantially consists of first speech formant frequencies and a second frequency range that substantially consists of second formant frequencies of the audio signal;	differentially amplifying a first frequency range that substantially comprises first speech formant frequencies and a second frequency range that substantially comprises second formant frequencies of the audio signal;	The current claim is encompassed by the previous claim, as the inclusive language "substantially consists" is a qualified variation of "comprises"
mixing an injected inaudible signal tone with the audio signal;	mixing an injected inaudible signal tone with the audio signal;	These limitations are identical.
sensing a level of the signal tone within the audio signal received; and	sensing a level of presence of the signal tone; and	The current claim is encompassed by the previous claim, as "presence" is understood to mean "level of the signal tone within the audio signal received."
controlling a gain for amplification of the second frequency range based on the level of the signal tone sensed;	automatically controlling gain amplification of only the second frequency range based on the sensed level of the signal tone;	The previous claim provides a narrower scope in specifying "automatically controlling" instead of the current claim which does not specify this and thus encompasses it.
the controlling the gain for the amplification to be based on the level sensed, to substantially prevent regenerative oscillation of the audio signal and to amplify the second formant frequencies without creating howling.	the controlling the gain amplification based on the sensed level to substantially prevent regenerative oscillation of the audio signal and to amplify the second formant frequencies without creating howling.	Though not identical, these limitations are directed to the same subject matter.

Furthermore, there is no apparent reason why applicant was prevented from presenting claims corresponding to those of the instant application during prosecution of the application which matured into a patent. See *In re Schneller*, 397 F.2d 350, 158 USPQ 210 (CCPA 1968). See also MPEP § 804.

Conclusion

9. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

- Fischer (US Patent 4,332,979) teaches an electronic environmental acoustic simulator.
- Gambacurta, et al. (US Patent 4,939,782) teaches a self-compensating multiband graphic equalizer.
- Field et al. (US Patent 5,001,757) teaches an FM Tone Detector.
- Leveque (US Patent 5,095, 539) teaches a system and method of tone control using injected signal tones.
- Magotra et al. (US Patent 5,608,803) teaches a programmable digital listening system utilizing digital filters.
- Hall (US Patent 6,307,945) teaches a radio-based hearing aid system using frequency modulation methods.
- Benade (*Horns, Strings & Harmony*, Anchor Books, 1960.) teaches the elementary physics of resonant musical devices including horns.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to David Kovacek whose telephone number is (571) 270-3135. The examiner can normally be reached on M-F 9:00am - 5:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, David Hudspeth can be reached on (571) 272-7843. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

DMK 12/12/2007

DAVID HUDSPETH
SUPERVISORY PATENT EXAMINER
TECHNOLOGY CENTER 2600

Substitute for form 1449A/PTO **INFORMATION DISCLOSURE STATEMENT BY APPLICANT FORM PTO-1449 (Modified)** *(USE AS MANY SHEETS AS NECESSARY)*	***Complete If Known***	
	Application Number	Unknown
	Filing Date	October 27, 2003
	First Named Inventor	Gillray L. Kandell
	Art Unit	
	Examiner Name	
SHEET 1 OF 2	Attorney Docket Number	0200.002.01/US

U.S. PATENT DOCUMENTS

Examiner Initials*	Cite No.1	Document Number Number- Kind Code (*If known*)	Publication Date MM-DD-YYYY	Name of Patentee or Applicant of Cited Document	Pages, Columns, Lines, Where Relevant Passages or Relevant Figures Appear
/DMK/		US 3,763,333	10-1973	Lichowsky	
/DMK/		US 3,894,194	07-1975	Kryter	
/DMK/		US 3,967,067	06-1976	Potter	
/DMK/		US 4,051,331	09-1977	Strong et al	
/DMK/		US 4,099,035	07-1978	Yanick	
/DMK/		US 4,109,116	08-1978	Victoreen	
/DMK/		US 4,405,831	09-1983	Michelson	
/DMK/		US 4,517,415	05-1985	Laurence	
/DMK/		US 4,633,498	12-1986	Warnke et al.	
/DMK/		US 4,731,850	03-1988	Levitt et al.	
/DMK/		US 4,852,175	07-1989	Kates	
/DMK/		US 4,879,749	11-1989	Levitt et al.	
/DMK/		US 4,947,432	08-1990	Topholm	
/DMK/		US 4,985,925	01-1991	Langberg et al.	
/DMK/		US 5,003,606	03-1991	Bordewijk	
/DMK/		US 5,033,090	07-1991	Weinrich	
/DMK/		US 5,271,397	12-1993	Seligmann et al.	
/DMK/		US 5,285,502	02-1994	Walton et al.	
/DMK/		US 5,303,306	04-1994	Brillhart et al.	
/DMK/		US 5,343,532	08-1994	Shugart, III	
/DMK/		US 5,347,584	09-1994	Narisawa	
/DMK/		US 5,388,185	02-1995	Terry	
/DMK/		US 5,420,930	05-1995	Shugart, III	
/DMK/		US 5,506,910	04-1996	Miller	
/DMK/		US 5,594,387	01-1997	Kagawa	
/DMK/		US 5,621,802	04-1997	Harjani	
/DMK/		US 5,966,400	10-1999	Den Braber	
/DMK/		US 6,128,369	10-2000	Bowker	

OTHER PRIOR ART -- NON PATENT LITERATURE DOCUMENTS			
Examiner Initials*	Cite No.1	Include name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc.), date, page(s), volume-issue number(s), publisher, city and/or country where published.	T2
/DMK/		Maxwell, jOseph A. and Zurek, Patrick M., *Reducing Acoustic Feedback in Hearing Aids,* IEEE Transactions on Speech and Audio Processing, vol. 3, No. 4, Jul. 1995, pp. 304-313.	
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ACOUSTIC SYSTEMS IN BIOLOGY

Neville H. Fletcher
CSIRO Australia
and Australian National University

New York Oxford
OXFORD UNIVERSITY PRESS
1992

Oxford University Press

Oxford New York Toronto
Delhi Bombay Calcutta Madras Karachi
Kuala Lumpur Singapore Hong Kong Tokyo
Nairobi Dar es Salaam Cape Town
Melbourne Auckland

and associated companies in
Berlin Ibadan

Published by Oxford University Press, Inc,
200 Madison Avenue, New York, New York 10016

Oxford is a registered trademark of Oxford University Press

Library of Congress Cataloging-in-Publication Data
Fletcher, Neville H. (Neville Horner)
Acoustic systems in biology / by Neville H. Fletcher.
p. cm. Includes bibliographical references and index.
ISBN 0-19-506940-4
1. Bioacoustics. I. Title
QP461.F52 1992 591.19' 14—dc20 91-37413

10 PIPES AND HORNS

SYNOPSIS. Pipes and horns are important components of many auditory and vocal systems, and their dimensions are often comparable with or larger than the wavelength of the sounds with which they interact. It is wrong to simply apply naive ideas of "open" or "stopped" pipes when analyzing such systems, for only occasionally are these approximations valid. More generally we must consider the waves flowing in both directions in the pipe or horn and use a set of four impedance coefficients Z_{ij}, as shown in Fig. 10.1, to represent the element by an electric analog network. The values of these impedance coefficients can be readily calculated in terms of the dimensions of the pipe or horn and the operating frequency. In narrow pipes, as shown in Fig. 10.2, the speed of sound may be somewhat reduced, and the attenuation α can become quite high.

To illustrate the behavior of pipes in acoustic systems, Fig. 10.4 shows the results of a calculation of the resonance frequencies of a pipe terminated by a cavity, as shown in Fig. 10.3, as the cavity volume is increased. The lowest resonance frequency moves from the stopped-pipe value to a much lower Helmholtz frequency, while each upper resonance shifts from the stopped-pipe to the next-lower open-pipe value. In biological systems, pipes are rarely either rigidly stopped or fully open.

Horns are symmetrical flaring structures as shown in Fig. 10.5. For the rather short horns found in biological systems, the exact horn profile has only a minor influence on acoustic behavior, and we must analyze the whole system to determine its response. It is instructive, however, to calculate the pressure gain in the throat of a horn, blocked by a very high impedance, for a plane wave falling normally on its mouth. The results, for three particular horns having the same length and the same throat and mouth diameters but flare profiles of parabolic, conical, or exponential shape, are given in Fig. 10.6. All horns have high pressure gain in a limited pass-band between upper and lower cutoff frequencies. The lower cutoff frequency is nearly the same for all horns and depends on the mouth and throat diameters and the length. The upper cutoff frequency varies more and is influenced by the horn profile. The width of the pass band increases as the length of the horn increases. The effects of terminating the horn with a matching load or with a resonant diaphragm are shown in Fig. 10.7. In both cases the gain is severely degraded, showing that the termination must be taken into account when considering total system behavior.

It is interesting to consider the behavior of an obliquely truncated horn, as shown in Fig. 10.8, since this bears some resemblance to an animal pinna. The direction of maximum acoustic sensitivity varies somewhat with frequency, as shown in Fig. 10.9, and the long oblique flap adds appreciably to the gain.

At very high frequencies, the behavior of a horn becomes complicated, and higher modes, such as illustrated in Fig. 10.10, can propagate, though they are quickly attenuated when the horn diameter narrows to less than the sound wavelength. A horn the shape of a shallow cylinder with an off-axis coupling to the auditory system, as illustrated in Fig. 10.11(a), constitutes a simple model for the human pinna. Obliquely incident sound can excite both the plane-wave mode and the first antisymmetric mode, at appropriate resonance frequencies for each, and the auditory system will be driven by the sum of these two pressures, while normally incident sound can excite only the plane-wave mode. The horn gain will thus have a different frequency dependence for different angles of incidence, as illustrated in Fig. 10.11(b).

In the region where the horn diameter is much greater than the sound wavelength, it

is appropriate to use an optical analog for wave propagation, with reflection from the sides of the horn. Because such a reflector cannot focus the sound into a region smaller than a wavelength in diameter, it is advantageous to use the reflector to focus sound onto the mouth of a smaller horn, as shown in Fig. 10.12, the restriction being that the horn mouth must be at least one wavelength in diameter for optimum efficiency. This appears to be an appropriate model for the pinnae of animals such as bats.

10.1 Acoustic Elements

Two or the most important elements of acoustic systems in biology are pipes and horns, as exemplified by the trachea in the vocal system, and by the meatus and the external pinna in auditory systems. In nearly all cases in which they occur, the dimensions of these structures are comparable with the wavelength of sound, at least for the higher frequencies with which the system must deal. Their exact shapes and dimensions therefore become significant, and we need a detailed discussion to elucidate their behavior. This more detailed treatment must, of course, reduce to the simpler version presented in Chapter 8 at frequencies low enough that the wavelength greatly exceeds the component dimensions.

The analysis of these extended systems can be made arbitrarily complex, by including refinements of geometry and material properties, but we shall be content with a simple treatment because we are concerned with generalities of behavior, rather than with fine details. The principal extension necessary to the electric analog theory developed in Chapter 8 is the recognition that pipes and horns have two ends, and that the acoustic flow in at one end is not necessarily equal to the flow out at the other. We therefore need a more general circuit element to describe their behavior and to serve as an analog in our electrical networks. This will emerge naturally as we proceed.

10.2 Pipes and Tubes

For our purposes there is no real distinction between a pipe and a tube, so we shall generally use the term pipe to mean any long duct with constant cross section. It turns out that the behavior of such a pipe does not depend greatly upon the shape of its cross-section, and is not much affected if it is bent, provided that the bend is not too sharp on the scale of the pipe diameter. An initial treatment in terms of long straight pipes can therefore be readily generalized.

It would be possible to develop our discussion in terms of the normal modes of pipes, in much the same way as we did for strings, but there is a major disadvantage to this approach. It works well for strings because they are generally anchored to a structure of relatively high rigidity, so that the boundary conditions at the two ends of the string are well defined. Simple treatments of pipes do the same thing, and discuss "open" and "stopped" pipes and their normal modes. In biological systems, however, the boundary conditions are almost never as simple as this. An "open" end is often either partly obstructed by a protective flap, or terminates in a horn, while a "stopped" end is usually blocked by a tympanum or

similar structure that is far from rigid, or terminates in a cavity. Many discussions of auditory and vocal systems are marred by simplistic application of these concepts. Instead, therefore, we develop our discussion in terms of waves in the pipe. This immediately gives greater generality, but allows recovery of the simple ideas when the boundary conditions are appropriate.

Suppose we have a pipe of cross-section S extending in the x direction from $x = 0$ to $x = l$. If we confine our attention to the behavior of the system at a fixed frequency ω, then the most general possible situation reduces to a wave with wave number $k = \omega/c$ and amplitude A (the complex quantity including a phase factor) traveling in the $+x$ direction, together with a similar wave of amplitude B traveling in the $-x$ direction. The acoustic pressure $p(x)$ at any point x is then simply the sum of the contributions from these two waves, so that

$$p(x) = Ae^{-jkx} + Be^{jkx} \tag{10.1}$$

where we have dropped the time factor $e^{j\omega t}$ for convenience. Note that the minus sign in the exponent refers to the wave traveling in the $+x$ direction, as discussed in Section 6.2. Now a wave in a pipe propagates in just the same way as a plane wave in free space, if we neglect the very small viscous friction at the walls, so that the acoustic particle velocity in the direction of propagation is $v = p/\rho c$ as we saw in (6.17). Because the cross-section area is S, this means that the acoustic volume flow associated with a wave in the pipe is $U = pS/\rho c$. Adding the contributions of the two waves in (10.1) and taking note of their opposite propagation directions then gives

$$U(x) = (S/\rho c)(Ae^{-jkx} - Be^{jkx}) . \tag{10.2}$$

It is convenient to define the quantity

$$Z_0 = \rho c/S \tag{10.3}$$

known as the characteristic impedance of the pipe. From (10.1) and (10.2) it is seen to be the input impedance p/U for an infinitely long pipe with no returning wave.

To describe the acoustic volume flow in the pipe in a symmetrical manner, we write the flow into the pipe at $x = 0$ as U_1 and the flow into the pipe at the other end $x = l$ as U_2, with the pressures similarly labeled as in Fig. 10.1. Then we can write the relation between these quantities as

$$p_1 = Z_{11}U_1 + Z_{12}U_2$$
$$p_2 = Z_{21}U_1 + Z_{22}U_2 \tag{10.4}$$

where the complex quantities Z_{ij} are impedances of some sort. Note that the flows and pressures are arranged symmetrically in both acoustic and electric analogs. If, for some reason, the direction of one flow or the sense of one pressure or voltage is opposite to that shown, then it should carry a minus sign with it. Of course, if we measure everything in the reverse sense, corresponding to turning the diagram upside-down, then all the minus signs cancel.

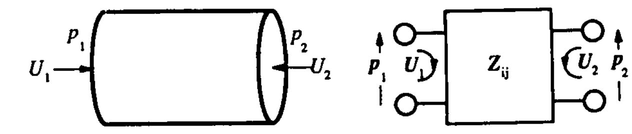


Figure 10.1 (a) Acoustic flows U_i and pressures p_i for a length of pipe; (b) the corresponding electric analog element.

A little algebraic manipulation of (10.4) using (10.1) and (10.2) then leads to the results

$$Z_{11} = Z_{22} = -jZ_0 \cot kl$$
$$Z_{12} = Z_{21} = -jZ_0 \operatorname{cosec} kl . \tag{10.5}$$

The equalities $Z_{11} = Z_{22}$ and $Z_{21} = Z_{12}$ clearly follow from the symmetry of the problem, since the pipe looks exactly the same from both ends. The fact that $Z_{21} = Z_{12}$ is an expression of the reciprocity theorem, which we have mentioned several times before, and is actually true even for systems, such as horns, that are not symmetrical.

It is easy to see how we can deduce the normal mode frequencies for ideally open and stopped pipes from these results. For a pipe ideally open at both ends, $p_1 = p_2 = 0$. A little algebra then shows, from (10.4) and (10.5), that this requires that $\cos^2 kl = 1$, which means that $kl = n\pi$ or $\omega_n = n\pi c/l$, with $n = 1, 2, 3, \ldots$. For a pipe open at end 1 and stopped at end 2, $p_1 = 0$ and $U_2 = 0$ so that, from the first equation of (10.4) and (10.5), $\cot kl = 0$ or $kl = (n - {}^1/_2)\pi$ and $\omega_n = (n - {}^1/_2)\pi c/l$.

Of rather more importance is the input impedance $Z_{IN} = p_1/U_1$ for open and stopped pipes. A little algebra immediately gives the results

$$Z_{IN}^{open} = jZ_0 \tan kl \rightarrow \frac{j\omega l}{S} \tag{10.6}$$

$$Z_{IN}^{stopped} = -jZ_0 \cot kl \rightarrow \frac{\rho c^2}{j\omega l S} . \tag{10.7}$$

The final form of writing in each case is for the low-frequency limit in which $kl \ll 1$. Clearly the results are equivalent to those given in Table 8.1 for the lumped-

component low-frequency approximation, the stopped pipe behaving like an enclosure of volume $V = lS$.

From the trigonometric forms of the impedance coefficients Z_{ij} given in (10.5), it is clear that their behavior with frequency is complicated. Their values go through a sequence of infinities and zeros as the frequency is increased, and these are related to potential resonances of the system. The actual resonances may not occur at these singular points, but may be displaced because of the impedance of other acoustic components connected to the two ends of the pipe. We look at an example of this in Section 10.4.

10.3 Wall Losses

We have already noted in Section 8.4 that we should make allowance for viscous and thermal losses to the walls of narrow pipes. These corrections become relatively less important at high frequencies, but they can have very significant influence on the behavior of an acoustic system when the pipe involved is very narrow, as may be the case in insect auditory systems. The exact results are a little complicated, but essentially what happens is that the wave number k should be modified from its simple value ω/c to become a complex quantity

$$k = \frac{\omega}{c'} - j\alpha . \qquad (10.8)$$

The propagation speed c' of the wave in the pipe is reduced somewhat below its open-air value c, and the wave amplitude attenuates with distance as $e^{-\alpha x}$. The behavior of c' and α as functions of tube radius and frequency is shown in Fig. 10.2. We can usually neglect the change in wave speed, except in the very narrow pipes of insect systems, but the attenuation losses may be important in other systems. For pipes more than about 1 mm in diameter it is an adequate approximation to take

$$\alpha \approx 1.2 \times 10^{-5} \omega^{1/2} a^{-1} . \qquad (10.9)$$

For very narrow pipes, the numerical coefficient in this equation is approximately doubled.

When it comes to taking account of these wall losses in the behavior of the pipe, the formal procedure is simply to substitute the complex value of k into the relations (10.5). As discussed in Appendix A, trigonometric functions with complex arguments can be simply expanded in terms of the hyperbolic functions sinh and cosh, which are readily evaluated. The formal generalizations of the expressions given in (10.5) are set out in Appendix B. In analyzing the behavior of a system, as we shall see in the next chapter, we do not need to do this until we write the final computer program, and then it is just a case of being careful with the algebra.

An extreme case of wall interaction can occur if the wall is actually porous,

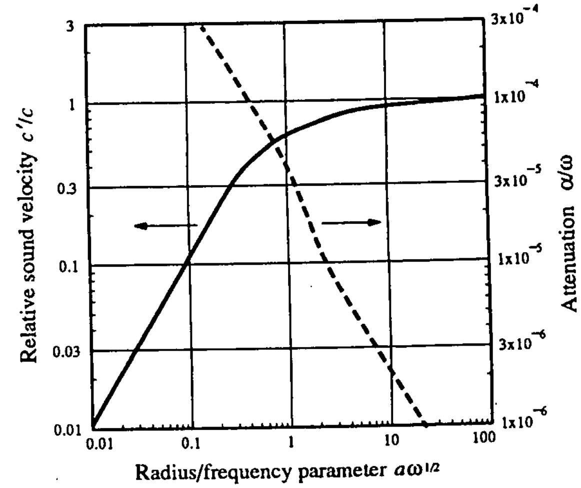


Figure 10.2 Sound speed c' and attenuation coefficient α in a pipe of radius a at angular frequency ω. The parameter $a\omega^{1/2}$ is 1.7 times the ratio of the pipe radius to the viscous boundary layer thickness.

as may be the case in some avian auditory systems. We shall not examine this in great detail, but a quick calculation shows the possible magnitude of the effect. If L is the acoustic inertance per unit length of pipe and C the acoustic compliance per unit length, then the sound propagation speed in the pipe is $c' \approx (LC)^{-1/2}$ and the characteristic impedance of the pipe is $Z_0' = (L/C)^{1/2}$. The inertance is contributed only by the mass of air in the free part of the pipe, of cross section S, while the compliance is contributed by the whole cross section, including the air permeating the porous walls. Suppose that the effective cross-section of pipe including the entire porous area is S'. Simple substitution of $L = \rho/S$ and $C = S'/\rho c^2$ then shows that the propagation speed and characteristic impedance in such a pipe are

$$c' \approx c(S/S')^{1/2} \qquad Z_0' \approx Z_0(S/S')^{1/2} . \qquad (10.10)$$

For a relatively narrow auditory canal connecting two simple ears, it is not hard to contemplate a situation in which S' is several times S, so that the effective speed of sound within the auditory canal is perhaps only half that in the free air. We should also expect significant attenuation due to motion of the air into the pores of the wall.

10.4 Helmholtz Resonator

In Section 8.6 we discussed the simple Helmholtz resonator, which consists of an enclosed volume vented through a pipe, using low-frequency approximations. Let

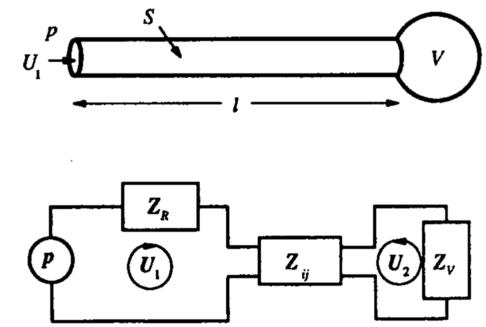


Figure 10.3 A Helmholtz resonator, together with its electric analog circuit. The directions of the circulating currents U_1 and U_2 are chosen, for convenience, so that they enter the two-port pipe analog symmetrically.

us now repeat the analysis with a better treatment of the pipe. There are two reasons for doing this. The first is to illustrate how we should use the impedance coefficients Z_{ij} when solving analog networks, and the second is to show the fallacy of treating pipes as "open" or "stopped" in real acoustic systems.

Figure 10.3 shows the Helmholtz resonator together with its analog network. We have assumed that the enclosed cavity of the resonator is small enough that it can be treated as a simple compliance, though a more complex treatment is, of course, possible. In drawing the currents in the network, we have to be careful that they enter the pipe element Z_{ij} in the same sense at each end, otherwise some of the signs will need to be changed. The network equations are simply

$$(Z_R + Z_{11})U_1 + Z_{12}U_2 = p \tag{10.11}$$

$$Z_{21}U_1 + (Z_V + Z_{22})U_2 = 0 \tag{10.12}$$

and these can be solved to give the flow U_1 through the neck as

$$U_1 = \frac{(Z_V + Z_{22})p}{(Z_R + Z_{11})(Z_V + Z_{22}) - Z_{12}Z_{21}}. \tag{10.13}$$

The resonance frequencies are given by the minima of the denominator.

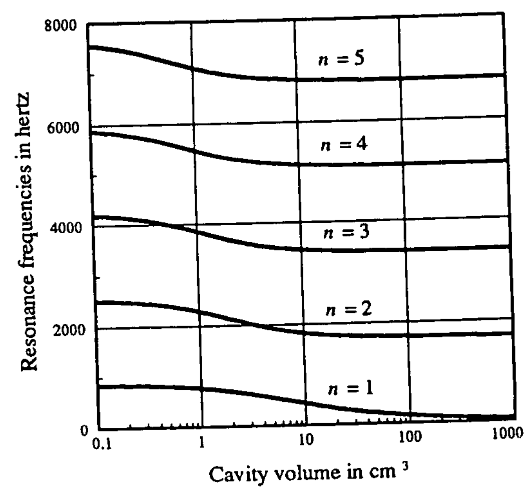
For the simple problem of resonance frequencies we are examining here, we can neglect Z_R in comparison with Z_{11} and ignore the resistive parts of all the other impedance coefficients. The denominator is then entirely real, and it vanishes at the resonance frequencies. If we take V to be the volume of the cavity so that $Z_V = -j\rho c^2/V\omega$, and use the explicit expressions for the Z_{ij} given in (10.5), we find the resonance condition to be

$$\frac{\omega V}{Sc}\tan\frac{\omega l}{c} = 1. \tag{10.14}$$

If the length of the tube is much less than the sound wavelength, so that $\omega l/c \ll 1$, we can take the tan function to be approximately equal to its argument, so that (10.14) reduces to the value found before for the Helmholtz frequency, $\omega_H = c(S/lV)^{1/2}$. More generally, however, (10.14) has many possible solutions, and it is informative to see how these resonance frequencies vary as we change the volume of the terminating cavity. This is illustrated in Fig. 10.4. When the cavity volume is very small, the resonance frequencies of the system are $\omega_n = (n - {}^1/_2)\pi c/l$, which are the resonance frequencies of a stopped pipe of length l. As the cavity volume increases, all the resonance frequencies fall progressively. The lowest resonance, for $n = 1$, tends to the Helmholtz resonance frequency for the system, while the higher resonances with $n > 1$ tend to the resonances of an open pipe, though with the index n reduced by 1, so that $\omega_n = (n-1)\pi l/c$.

The lesson to be learned from this analysis applies to many biological systems, whether the pipe terminates in a cavity, a tympanum, or some other kind of impedance. That lesson is that the nature and magnitude of this terminating impedance has a very significant effect on the resonance frequencies of the pipe,

Figure 10.4 Resonance frequencies for the first five modes of the Helmholtz resonator of Fig. 10.3 as a function of the cavity volume V. The pipe length l is 10 cm and the pipe diameter 10 mm. For very small volumes, the resonance frequencies are those of a stopped pipe, while for large volumes they are those of an open pipe, supplemented by a very low-frequency Helmholtz resonance. The assumptions underlying the model cease to be valid for very large cavities and high frequencies.



particularly those of the lower resonances. A very similar conclusion comes from a study of the resonance frequencies of a pipe closed with a light slack diaphragm, as set out in the discussion examples.

Before leaving the subject of this section we must insert a cautionary remark. It might seem that our discussion of the Helmholtz resonator is now exact, and that all higher modes have been properly included, but this is not true. We have progressed from the simple single-mode model of Section 8.6 to a much more sophisticated model, it is true, but our treatment of the behavior of the cavity is still oversimplified. It is valid to treat the cavity as a lumped acoustic compliance, as we have done here, only so long as its dimensions are small compared with the sound wavelength involved. This is appropriate for the first few modes while the cavity remains small compared with the pipe length, but is certainly not true for higher modes. Our next model in the series must therefore include the modes of the cavity itself. We do not need to do this for any of our biological problems, but it is a good example of the use of an appropriately complex model at each stage of the discussion.

10.5 Simple Horns

A striking feature of many mammalian auditory systems is the pair of large horn-like pinnae mounted on the sides of the head. It is qualitatively clear that they act in some way to collect sound energy and funnel it to the tympanum, but we require some analysis to evaluate just how this occurs and to explore the frequency response and directionality of the horn. The horns of typical pinnae are not truncated at right angles to their axis, but rather at an oblique angle. We shall return to consider this complication in the next section after we have understood the behavior of normally truncated horns.

It is traditional in acoustics texts to investigate the behavior of horns of infinite length and having one of a number of idealized geometric profiles—conical, exponential, "hypex" (Salmon), "Bessel", etc. The results are elegant, but are of little use in the present context because biological horns are necessarily finite in length and often quite short, and their geometry rarely conforms closely to any of these idealized types. We shall therefore consider only horns of finite length, and examine the behavior of just three types—a conical horn, a horn with a more rapid and quasi-exponential flare, and a horn with a slower and quasi-parabolic flare—as illustrated in Fig. 10.5. Between them they serve as models for all the horns encountered in biological systems.

The wave equation for propagation in a horn with cross section $S(x)$ is the so-called horn equation or Webster equation

$$\frac{\partial^2 p}{\partial t^2} = \frac{c^2}{S}\frac{\partial}{\partial x}\left(S\frac{\partial p}{\partial x}\right). \qquad (10.15)$$

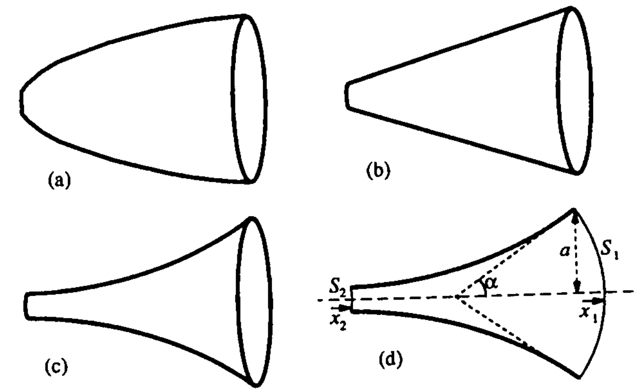


Figure 10.5 (a)–(c) Three horn shapes, all with the same mouth and throat dimensions and the same length. The horn profiles are (a) parabolic, (b) conical, and (c) exponential. Diagram (d) defines the coordinates and illustrates the curvature of the wavefront at the mouth of the horn, together with the associated tangent cone.

Clearly it reduces to the ordinary wave equation (6.8) when S is constant, as in a cylindrical pipe. It is not an exact equation, particularly at high frequencies and for large flare rates on the horn, but is a useful approximation. The explicit forms for the profiles of the horns we shall consider are

$$S(x) = S_0 x^2 \qquad \text{(conical)} \qquad (10.16)$$

$$S(x) = S_0 e^{2mx} \qquad \text{(exponential)} \qquad (10.17)$$

$$S(x) = S_0 x \qquad \text{(parabolic)} \qquad (10.18)$$

The parabolic and conical horns are both members of a more general family with $S(x) = S_0 x^m$. These are called Bessel horns because the propagating waves within them can be written quite generally in terms of Bessel functions.

Evaluation of the coefficients $\mathbf{Z}_{ij}$ for a finite horn follows just the same lines as the method used in the previous section for a cylindrical pipe. The one difference is that, instead of simple exponential behavior e^{-jkx} describing the propagating waves, we must use rather more complicated functions. For the conical horn, for example, the propagating wave has simple spherical form, so that its pressure and velocity components have the spherical-wave forms given in (6.33) and (6.35). For parabolic and exponential horns the behavior is a little more complicated. We need not go into details of the algebra involved, and it will suffice to quote the final results, which are collected in Appendix B for convenience. In each case we take the radius of the large end, or mouth, to be a, so that the mouth area is $S_1 = \pi a^2$,

and the radius of the small end, or throat, to be b, so that the throat area is $S_2 = \pi b^2$. The length of the horn between these ends is taken to be l.

The coefficients $\mathbf{Z}_{11}$ and $\mathbf{Z}_{12}$ thus apply to the horn viewed from its mouth, and the coefficients $\mathbf{Z}_{22}$ and $\mathbf{Z}_{21}$ to the same horn viewed from its throat. Because the horn looks different from its two ends, it is no surprise that $\mathbf{Z}_{22} \neq \mathbf{Z}_{11}$. However, when we work out expressions for $\mathbf{Z}_{12}$ and $\mathbf{Z}_{21}$, we find that $\mathbf{Z}_{21} = \mathbf{Z}_{12}$. We have remarked on this somewhat unexpected result before, in Section 8.3, in relation to levers, which are similarly asymmetric. It is a general property of linear systems without steady magnetic fields, and is called the reciprocity theorem.

The expressions for the $\mathbf{Z}_{ij}$ are, in each case, moderately complicated, but straightforward to evaluate. For all horns, the impedance coefficients are pure imaginary, as for the cylindrical pipe, if wall losses are neglected. For the exponential horn, and indeed for hypex horns more generally, there is a discontinuity in the mathematical form of the impedance coefficients at the frequency for which $k = m$, where m is the flare constant defined by (10.17). Below this frequency, called the flare cutoff frequency, we have only an exponentially attenuated disturbance, in-phase at all points, instead of propagating waves. This behavior blocks propagation completely in horns of infinite length, but its effect is much less severe in short horns. Indeed, the behavior of short exponential horns in this respect is very little different from that of horns of other profiles.

When we come to look more carefully at the propagation of waves in a horn, we find that the wavefronts are not plane, but rather curved, as indeed we have recognized in the case of the conical horn by using the spherical wave propagation equation. This means that the cross-section area $S(x)$ in the horn equation (10.15) is actually the area of the curved wavefront meeting the axis at position x, as shown in Fig. 10.5(d), rather than the geometric cross-section of the horn. Provided the horn does not flare too widely—in which case some of the other assumptions underlying the horn equation (10.15) also cease to hold—this does not make a great deal of difference to propagation within the horn. It does, however, have a significant effect at the mouth of the horn, where an incoming plane wave must make the transition to a curved wavefront in the horn. The incoming plane wave in fact drives different parts of the curved wavefront in different phase, so that the transformation of energy is incomplete. We need not go into the analysis of this effect in detail, and the result is relatively straightforward. Suppose that the radius of the horn mouth is a and the flare angle at the mouth—the semi-angle of a cone that would be tangent to the horn surface at its mouth, as shown in Fig. 10.5(d)—is α. Then the curvature mismatch reduces the coupling between a wave in the horn and an incident or outgoing plane wave by a factor

$$F_\alpha(ka) = \frac{\sin[(ka/2)\tan(\alpha/2)]}{(ka/2)\tan(\alpha/2)}. \tag{10.19}$$

This factor is unity at low frequencies, but reduces the efficiency of the horn at high frequencies. The coupling becomes zero when $ka = 2n\pi(\tan \alpha/2)^{-1}$.

In a complete treatment of horns it would be appropriate to make allowance for wall losses as we did for cylindrical pipes. The principle is the same, but exact calculation is difficult because the imaginary part of the wave number k varies along the horn as its radius changes. We could reasonably average the value of the reciprocal of the radius along the horn, since losses are proportional to this quantity, and then use the results (10.8) and (10.9) to assign an appropriate imaginary part to k. We shall not usually bother to do this, since in most auditory systems most of the damping is contributed by the tympanum and by radiation losses at the wide horn mouth. Only for long narrow horns, such as the trachea in the vertebrate vocal system or horns in certain insect auditory systems, are wall losses significant.

There is another effect that must be taken into account at high frequencies, and this arises from the fact that we are trying to build a simple one-dimensional model with a single acoustic current for a situation that is really three-dimensional. When we model a very small component in a sound field, there is no ambiguity about the sound pressure at the entry to the component; it is just equal to the sound pressure in the field. For an object of size comparable with or larger than the sound wavelength, however, the situation is more complex. We already know, for example, that the sound pressure close to a large plane baffle is twice the free-field sound pressure, because of the effect of the reflected wave.

Although it is difficult to analyze this situation in detail, we can arrive at the final result very simply. By a fundamental theorem in electric networks, Thévenin's theorem, any real generator can be replaced by an ideal generator in series with some appropriate impedance. Applying this to the pressure acting on the mouth of the horn, or of a pipe of cross-section S_1, we suppose the effective generator pressure to be $\boldsymbol{p}_E$ and the series impedance $\mathbf{Z}_E$. There are two steps in the calculation.

If we look at the case of an infinitely long pipe, parallel to the propagation direction and with walls that are infinitely thin and smooth, then the presence of this pipe makes no difference to the wave propagation. An analog network built up using the components of Fig. 8.4 shows a pressure generator $\boldsymbol{p}'$ feeding through a radiation transition $\mathbf{Z}_R$ to the input impedance $Z_0 = \rho c/S_1$ of the infinite pipe. If the flow into the pipe is to be the same as that in a plane wave, so that $\boldsymbol{U} = \boldsymbol{p}/Z_0$, then we must have an effective generator magnitude

$$\boldsymbol{p}' \approx \boldsymbol{p}(1 + \mathbf{Z}_R S_1/\rho c) \tag{10.20}$$

where $\mathbf{Z}_R$ is the radiation impedance of an aperture of area S_1 as given by (7.17) and Fig. 7.5. To a good approximation, if $S_1 = \pi a^2$, then $\mathbf{Z}_R S_1/\rho c \approx 0.6jka$ for $ka < 1.7$ and ≈ 1 for $ka > 1.7$. For small apertures, therefore, $\boldsymbol{p}' \approx \boldsymbol{p}$, while for large

apertures $p' \approx 2p$. The source impedance Z_E is just the radiation impedance Z_R, which is, of course, routinely included in our analog network anyway, so that we need only replace the free-field pressure p by the equivalent pressure p'.

In the case of a horn, we must also make allowance for the curvature of the wavefront at the horn mouth, as expressed by the function $F_\alpha(ka)$ of (10.19). This can be done simply by generalizing (10.20) to make the effective pressure source strength

$$p_E = p'F_\alpha(ka) = p(1 + Z_R S_1/\rho c)F_\alpha(ka)\,. \tag{10.21}$$

This expression is what we must use in our analog network for a horn excited by a plane wave parallel to its axis. We see in Section 10.6 what modification we should make when the sound is incident at an angle to the horn axis.

Although it is of only limited use to calculate the behavior of a horn as a separate component, since the acoustic impedance of whatever is connected to its throat has a large influence, nevertheless such a simple calculation is instructive. We therefore calculate the acoustic pressure p_T generated in the throat of a horn by a plane wave incident along the axis, for three simple cases. The first case is that of a stopped horn, in which the throat is rigidly blocked. A small probe microphone of very high acoustic impedance might then be inserted to measure the pressure. The second case is that of a matched horn, in which we suppose the throat to be connected to an infinite pipe of matching diameter. Again a small probe microphone could be used to measure the throat pressure. Thirdly, we suppose the throat to be terminated by a moderately thick and only lightly damped resonant membrane.

The network equations for a horn with an acoustic load Z_L at its throat are easily written down with the help of the analog network of Fig. 10.6. Note that the driving pressure is taken as p_E, in accord with our discussion above, and the internal impedance of the generator is represented by the radiation impedance Z_R at the mouth of the horn, which we always include anyway. These equations are

$$\begin{aligned} p_E &= (Z_R + Z_{11})U_1 + Z_{12}U_2 \\ 0 &= Z_{21}U_1 + (Z_L + Z_{22})U_2 \\ p_T &= Z_{21}U_1 + Z_{22}U_2 = -Z_L U_2. \end{aligned} \tag{10.22}$$

The third equation is not usually one of the set, and is very nearly the same as the second, but is divided up so as to give two equivalent expressions for the pressure p_T in the throat of the horn.

For the case of a horn with a rigidly stopped throat, $Z_L = \infty$, which means that $U_2 = 0$, either from physical considerations or from the second of equations (10.22). The first of (10.22) then gives U_1 and the first form of the third equation gives the throat pressure as

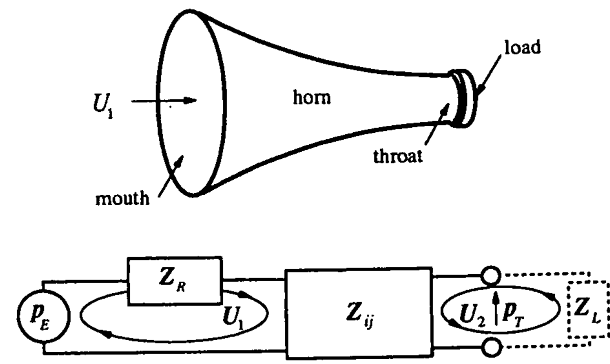


Figure 10.6 Electric network analog for a horn, loaded at its throat by an acoustic impedance Z_L. p_E is the analog driving pressure and Z_R the radiation impedance at the horn mouth.

$$p_T = \left[\frac{Z_{12}}{Z_R + Z_{11}}\right]p_E \tag{10.23}$$

with p_E given by (10.21). For the matched horn, we define the matching load at the throat to be $Z_2 = \rho c/S_2$, and the network equations are then

$$\begin{aligned} p_E &= (Z_{11} + Z_R)U_1 + Z_{12}U_2 \\ 0 &= Z_{12}U_1 + (Z_{22} + Z_2)U_2. \end{aligned} \tag{10.24}$$

Again, these are easily solved to obtain $p_T = -Z_2U_2$ in the form

$$p_T = \left[\frac{Z_2 Z_{12}}{(Z_{11} + Z_R)(Z_{22} + Z_2) - Z_{12}^2}\right]p_E\,. \tag{10.25}$$

The solution when the horn is terminated by a resonant diaphragm is formally the same as (10.25) if we replace the terminating impedance Z_2 by the impedance Z_T appropriate for a tympanum, as given in Table 8.1.

It is easy to calculate these responses as functions of frequency, using the explicit expressions given in Appendix B for the Z_{ij} for different types of horns. The calculation is simplified since we are interested only in the absolute value of the pressure, not its phase. The first and simplest case is that in which the throat of the horn is assumed to be rigidly stopped. Figure 10.7 shows the calculated gain $G = 20\log_{10}(p_T/p)$ in decibels for three horns, each having the same length, mouth and throat diameters, but being of parabolic, conical, or exponential profile respectively. The average flare angle is about 27° in each case, but the flare is differently distributed along the length of the horn. Physically these horns are

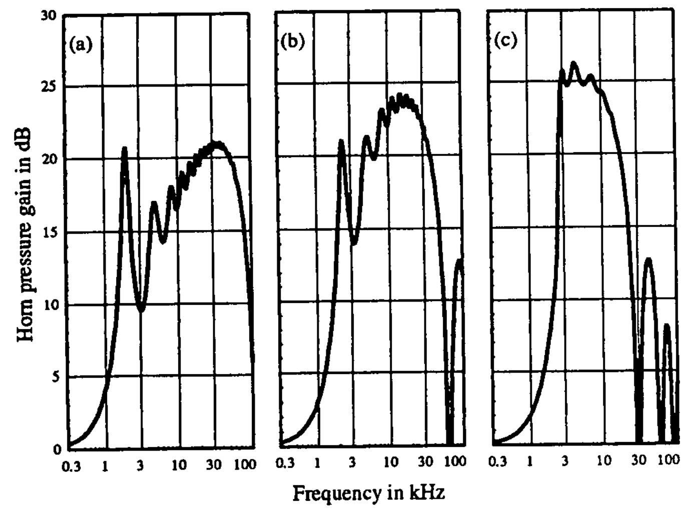


Figure 10.7 Response of (a) parabolic, (b) conical, and (c) exponential horns of the same length (50 mm), mouth diameter (50 mm), and throat diameter (5 mm), with the throat rigidly stopped.

about the size of the pinnae of a large mammal. The curves can be scaled to any absolute size of horn of the same shape, an increase in dimensions by a factor K changing the frequency scale by a factor $1/K$. These curves have been verified, at least for the case of a conical horn, up to a scaled frequency that includes the first subsidiary maximum above the cutoff. The fact that the simple theory works so well, even at high frequencies, arises from the fact that higher modes generated at the mouth, which we discuss in Section 10.8, are unable to propagate to the narrow throat. There is some doubt, however, about the calculation for the parabolic horn at high frequencies, because of focusing effects to be discussed later.

It is notable that the performance of all three horns is broadly similar. The gain falls to unity, or 0 dB, as the frequency falls below a lower cutoff which, for these particular horns, is at about 2 kHz. This corresponds to the frequency of the actual flare cutoff (2.5 kHz here) in the case of an exponential horn, but we can see that such behavior is quite general. There is then a pass band in which the gain is quite high, extending to an upper cutoff above which the gain once more becomes small, and may even be less than 0 dB in certain frequency intervals. The upper cutoff frequency varies somewhat with horn profile.

It is useful to have approximate formulae for estimating the frequencies of the lower and upper cutoff in an arbitrary horn, and these can be provided if the flare angle at the mouth is not too large. The lower cutoff frequency is given approximately by

$$\omega_{\text{lower}} \approx \frac{c}{l}\log_{10}\left(\frac{S_1}{S_2}\right) \tag{10.26}$$

and the upper cutoff approximately by

$$\omega_{\text{upper}} \approx \frac{2n\pi c}{a\tan\alpha/2}. \tag{10.27}$$

These results are accurate to within about ±30% for ordinary horn shapes. Clearly a more careful calculation is necessary if the result is important. Note that the pass band of a horn with given mouth and throat diameters can be extended at both low and high frequencies by making it longer, while its performance at high frequencies alone can be improved somewhat, at the expense of the lower parts of the passband, by modifying the profile towards parabolic. There is, however, a measure of overestimation in the parabolic calculation at high frequencies because of the neglect of reflection effects, which we discuss below.

Within the pass band, the gain reaches a peak value of about

$$G_{\text{stopped}} \approx 10\log_{10}(S_1/S_2) + A \quad \text{dB} \tag{10.28}$$

for the stopped exponential horn. The constant A comes from (10.20), for we wish to compare the pressure in the blocked throat of the horn with the pressure we would measure with the same microphone set up in front of the throat-blocking plate alone, or equivalently with the pressure measured by a microphone of the same diameter as the throat. $A \approx 6$ dB if the mouth diameter is large compared with the sound wavelength and the throat diameter small, but $A \approx 0$ dB if both are large or both small compared with the sound wavelength. Conical and parabolic horns do not achieve a gain quite as high as this.

In addition to the general shape of the stopped-throat horn gain characteristic, it is worthwhile to note the presence of resonance peaks, the first, at about 2 kHz for the particular horns in Fig. 10.7, being the most prominent. The extent to which the resonances stand out above the general curve depends upon the ratio of the length of the horn to its mouth diameter. If the horn is short and wide, then the radiation resistance at the mouth at the frequency of the first resonance is nearly $\rho c/S_1$ and the resonance Q is very low. This is usually the case for auditory pinnae, which have about the shape for which the figures were calculated. For long narrow horns, as found in musical instruments or in the mammalian vocal system, the first few resonances are much more pronounced. Note, incidentally, that the frequency of the first resonance, for a given horn length, depends appreciably upon the profile of the horn, as illustrated in Fig. 10.7.

Figure 10.8 shows the effect of the other two model terminations on the pressure gain of the conical horn. The effect on the gain of the other profiles is similar. When the horn throat is matched to an infinite-pipe load, the shape of the

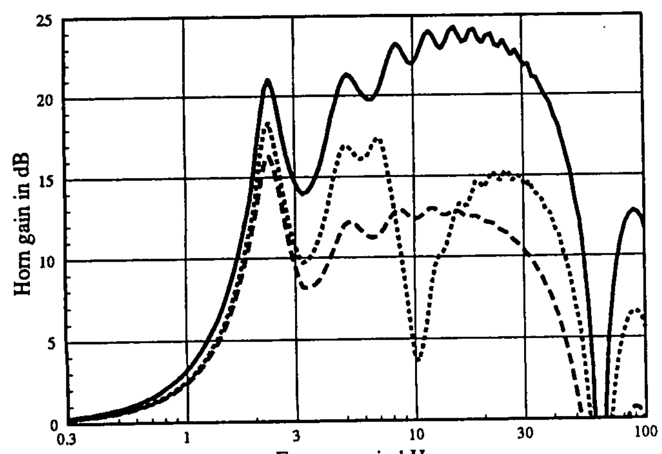


Figure 10.8 Response of the conical horn of Fig. 10.6 with the throat rigidly blocked (full curve), matched to an infinite-pipe load (broken curve), and terminated by a resonant diaphragm (dotted curve). The diaphragm has a thickness of 65 μm, a resonance frequency of 10 kHz, and a Q of 5.

gain curve is very little altered, but the actual pressure gain, measured in decibels, is approximately halved, representing about a 10 dB loss in performance. In the case of the resonant-membrane termination, there is a very marked decrease in pressure gain near the frequency of the diaphragm resonance, but a less severe effect at other frequencies. These calculations emphasize that we need to consider the performance of the whole system, and cannot simply combine the performance of isolated components measured under different conditions.

10.6 Directionality of a Horn

In the frequency range below that corresponding to the upper cutoff frequency of the horn, the wavefront at the mouth is curved with a dome height that is less than about half a wavelength, so that it is a reasonable approximation to regard it as plane for the purposes of calculation. The angular distribution of radiation when the horn is used as a transmitter, or the angular distribution of sensitivity when it is used as a receiver, is therefore nearly the same as that of an open pipe of the same diameter, as shown in Fig. 7.6. The angular response narrows as the frequency increases, the angular displacement of the −3 dB points being about

$$\theta_{-3\,\mathrm{dB}} \approx \pm \sin^{-1}(1.6/ka) \tag{10.29}$$

at least in the range $1 < ka < 10$. For $ka > 10$ wavefront curvature can no longer be neglected and the beam broadens and develops side lobes.

We can include this directional effect in the effective strength p_E of the network pressure generator, by generalizing (10.21) to include a directional factor $D_{ka}(\theta)$. The complete result for a wave of free-field pressure p incident at an angle θ to the axis of a horn with mouth area $S_1 = \pi a^2$ is then

$$p_E \approx p\left(1 + \frac{Z_R S_1}{\rho c}\right) F_\alpha(ka) D_{ka}(\theta) \,. \tag{10.30}$$

There is no simple expression for $D_{ka}(\theta)$, which is given by the curves of Fig. 7.6, but we note that $D_{ka}(0) = 1$. In the forward direction it is approximately equal to the corresponding expression for a baffled pipe,

$$D_{ka}(\theta) \approx 2J_1(ka \sin\theta)/ka \sin\theta \qquad \theta < \pi/2. \tag{10.31}$$

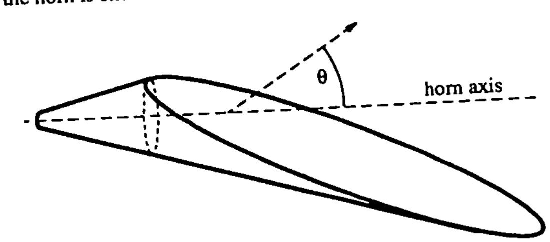
10.7 Obliquely Truncated Horns

It is a good first approximation to consider the external part of the pinna of many animals to have the form of an obliquely truncated simple horn, as shown in Fig. 10.9. The actual profile is often not far from conical, though our discussion in Section 10.5 leads us to expect that the exact flare shape is not critical.

Any moderately accurate treatment of such an obliquely truncated horn is very difficult, and even an approximate calculation such as the one referred to in the Bibliography [25] is quite complex and uncertain. The horn commences its flare with complete sides and then tapers away to an extended flap. We expect the normal part of the horn to behave simply, while the extended flap presumably both changes the direction of greatest sensitivity and adds somewhat to the acoustic gain.

Calculations confirm these expectations. For a typical upright pinna shape, as in a rabbit or a marsupial, the angle of greatest sensitivity, which we might call the acoustic axis, is shifted away from the horn axis in the direction of the open face of the pinna. The shift of the acoustic axis is frequency dependent, as shown

Figure 10.9 An obliquely truncated conical horn. The axis of maximum acoustic gain makes an angle θ with the geometric horn axis. The mouth of the "complete" part of the horn is shown dotted.



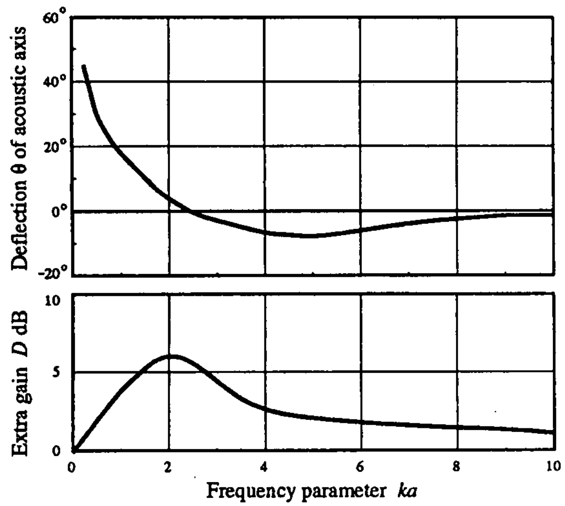


Figure 10.10 Angular displacement θ of the acoustic axis of an obliquely truncated conical horn with mouth radius a at the end of the complete part, and apex semi-angle 20°, truncated at an angle of 30°. The shape is about as shown in Fig. 10.9. Also plotted is the extra gain relative to a normally truncated horn the size of the complete part. (From [25].)

in Fig. 10.10, and it actually moves surprisingly far from the horn axis at very low frequencies. The importance of this shift at very low frequencies is, however, illusory, since the angular response is then extremely wide and the position of the acoustic axis hardly detectable. Of greater significance is the more moderate displacement of the axis at intermediate frequencies where the angular response is much sharper. We must be careful, however, about applying these ideas simplistically to real pinnae, in which the horn axis is often curved. This introduces reflection effects at high frequencies, as discussed in Section 10.9.

The extended curved flap of the pinna also adds to the gain of the horn, but only to a modest extent, the peak addition being about 6 dB for the case calculated. This too is frequency dependent, and the extra gain is added in a useful part of the pass band of the normal horn.

10.8 Higher Modes in Horns

As the frequency is increased, for a horn or pipe of given diameter, we ultimately reach a point where it is possible to have standing waves across the diameter. It is probably only near the mouth of auditory pinnae that this effect is significant, but it is worthy of a little comment, if not much detailed analysis.

Suppose we look carefully at the propagation of an acoustic wave of frequency

ω along a rather wide pipe of radius a. To analyze this properly we need to write down a full three-dimensional wave equation, since it is not clear, in a pictorial sense, that the wave will necessarily travel straight down the pipe, rather than moving at an angle and reflecting off the sides. Cylindrical polar coordinates (r, ϕ, z) are appropriate for this problem and, after some algebra, we find that the pressure in a general wave can be written as a sum of functions of the type

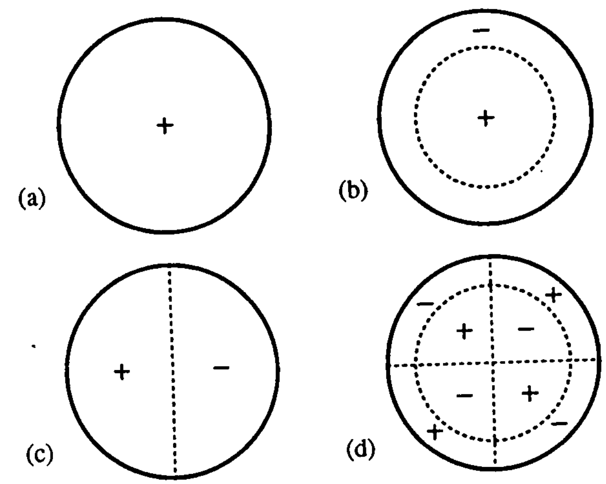
$$p_n(r,\phi,z) = A e^{-jkz} J_n(\kappa r) \sin n\phi \; e^{j\omega t} \tag{10.32}$$

where J_n is a Bessel function of order n. Bessel functions almost always turn up when we consider problems with circular symmetry, as we found also for the circular membrane. The axial wave number k is related to the frequency by

$$k^2 + \kappa^2 = (\omega/c)^2 \quad \text{or} \quad k = \sqrt{(\omega/c)^2 - \kappa^2} \tag{10.33}$$

and the allowed values of κ are those corresponding to standing waves across the pipe, for which $dJ_n(\kappa r)/dr = 0$ at the pipe wall $r = a$. If $n = 0$, the waves are axially symmetric and ϕ does not enter. The lowest mode then has $\kappa = 0$ and is the familiar plane wave. The next axially symmetric mode has $\kappa = 3.83/a$ and the cross-section shown in Fig. 10.11(b), with the central part moving in the opposite direction to the edge. If the frequency is low so that $\omega/c < \kappa$ then, from (10.33), k becomes

Figure 10.11 Patterns of acoustic pressure for the higher modes in a circular pipe or horn: (a) the plane-wave mode, (b) the second axially-symmetric mode, (c) the first antisymmetric mode, (d) a higher mode. Axial acoustic flow has opposite directions in regions marked + and −, and there is radial acoustic flow across the boundaries between these regions.



imaginary and the mode is exponentially damped with distance along the pipe rather than propagating. In wide pipes, however, the mode can propagate, its group velocity—the speed of propagation of a tone burst—being rather less than that of a plane wave, though the phase velocity ω/k is actually higher. Modes of this type are generated by a plane wave falling normally on the mouth of a horn, through the effects of wavefront curvature, but cannot propagate far towards its throat unless the frequency is very high. At very high frequencies additional modes of this type, with more annular regions of opposite motion, can be generated and propagate.

Perhaps more important are the modes with $n = 1$, which are antisymmetric across the pipe, one half of the air moving in antiphase with the other as shown in Fig. 10.11(c). Such waves are excited when a plane wave falls obliquely on the end of a horn, or quite generally in an obliquely truncated horn or pinna. The cutoff frequency for propagation of this mode is lower than for the first higher axial mode, because for it $\kappa = 1.84/a$. Again, the wave is sharply attenuated in the narrow part of a horn but, as we shall discuss in more detail in the next section, modes of this antisymmetric type may play a part in shallow pinnae such as found in primates and in humans. At much higher frequencies, modes with higher angular dependence $\sin n\phi$ and with several nodal circles, as shown in Fig. 10.11(d), can propagate.

It is useful to have an estimate of the rate of attenuation with distance for higher modes that cannot propagate. If the frequency is well below the propagation cutoff for the mode involved then, from (10.33), $k \approx j\kappa$ and the attenuation behavior is as $e^{-\kappa x}$. For the lowest asymmetric mode with $n = 1$, we have seen that $\kappa a = 1.84$, so that the amplitude of the mode is reduced by $e^{-3.7}$, or about 30 dB, over a distance equal to the diameter of the pipe. For higher modes the attenuation is even more rapid, so that only modes that are close to or above their propagation cutoff frequency have appreciable effect on the behavior of the system.

10.9 Shallow Asymmetric Horns

As an example of the importance of such higher horn modes in some auditory systems, consider the simple model shown in Fig. 10.12. This model has been used as a basis for analysis of the human external ear by Shaw (see the Bibliography). In this model the horn has become a simple short cylinder, with the throat a small aperture in its inner face, and we are concerned with the pressure driving flow into this throat. A plane wave incident at an angle to the axis of the cylinder can generate both a plane-wave mode and, if its frequency is high enough, modes with angular dependence $\sin(2n-1)\phi$, the first and most important of which is the antisymmetric mode with $n = 1$. Since the throat is placed off-center, it is affected by the pressure in this mode as well as by the pressure in the symmetric plane-wave mode.

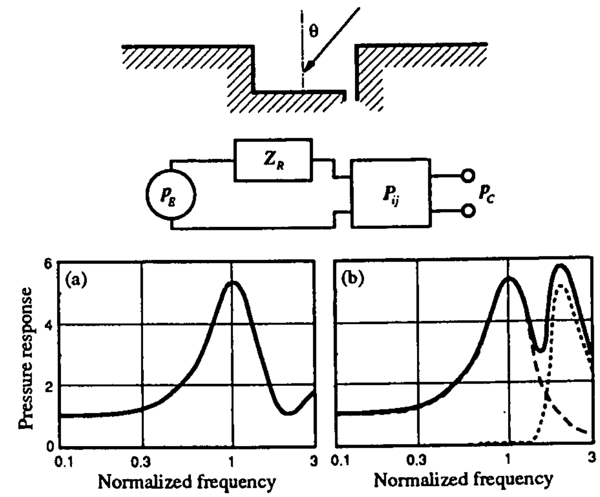


Figure 10.12 A simple model for an ear with an off-set auditory canal as examined experimentally by Shaw. The behavior can be analyzed with the help of the analog circuit shown, but a separate analog is required for the plane-wave and the antisymmetric modes. For sound incident at $\theta = 0$, only the plane-wave mode is excited, and the response at the entry to the auditory canal is as in (a). For oblique incidence ($\theta = 45°$) as in (b), the plane mode response is shown dashed, the antisymmetric mode dotted, and the total response as a full curve. (Adapted from [23].)

It is possible to calculate the response of the simple system shown in Fig. 10.12(a), measured at the auditory canal aperture, for sound incident at an arbitrary angle θ, but we shall not go through this exercise in detail. An outline shows the basis of the analysis and indicates the results in at least semiquantitative fashion. Shaw's investigation, incidentally, was experimental rather than theoretical. For simplicity we assume the impedance presented by the canal to be very high, so that it is effectively stopped, but we must repeat the warnings in Section 10.5 about interpretation of the calculated results!

If a is the radius of the cylindrical cavity and l its depth, then the plane-wave mode has a first resonance when the acoustic length $l' \approx l + 0.8a = \lambda/4$, the end correction being a little larger than usual because of the flange. If $l \approx a$, then this resonance occurs for $ka \approx 0.9$ or $\omega_+ \approx 0.9c/a$. The pressure will be a maximum at the inlet to the auditory canal at this frequency, and indeed since Fig. 7.5 shows that $R_R \approx 0.2Z_0$ for $ka = 0.9$, the pressure amplification will be about a factor 5, or 13 dB, at resonance.

It is easy to make this discussion quantitative, using the analog circuit shown in Fig. 10.12. Since there is no flow into the auditory canal, the pressure p_C at its entry is readily seen to be

$$p_C = \frac{P_{12}\,p_E}{Z_R + P_{11}} \tag{10.34}$$

where the generator pressure p_E is given by (10.30) and includes a factor $D_{ka}(\theta)$ taking account of the direction of incidence of the plane wave. The response for normal incidence ($\theta = 0$) is shown semiquantitatively in the graph of Fig. 10.12(a). For oblique incidence with $\theta = 45°$, the plane-wave mode gives the pressure contribution shown as a dashed curve in Fig. 10.12(b).

For the first resonance of the antisymmetric mode, we have already seen that the transverse wave number is $\kappa = 1.84/a$, and the axial wave number must still satisfy the requirement $ka \approx 0.9$, if we neglect possible modification of the end correction. From (10.33) we then find the resonance frequency $\omega_- \approx 2c/a$. Thus for the particular geometry considered, $\omega_- \approx 2\omega_+$. This antisymmetric resonance is rather less damped than the plane-wave resonance, because of the dipole nature of the motion, so the response peak is comparable in height to that of the plane-wave resonance. A significant difference arises, however, from the fact that the antisymmetric mode makes almost no contribution to the response below its cut-off frequency.

It is possible to formulate an expression very much like (10.34) for the pressure contribution of the antisymmetric wave, but this would take us beyond the scope of our discussion since we have not derived the appropriate pipe impedance coefficients for anything but the plane-wave mode. We can note, however, from (10.33), that P_{12} will be very small below the cut-off frequency of the cylindrical duct, so that the antisymmetric mode makes no contribution to the canal pressure below this frequency. We can also note that the expression in p_E taking account of incidence direction has a quite different angular dependence from the plane-wave case. Indeed for an antisymmetric wave $p_E = 0$ for normal incidence, and it goes through a maximum at an angle that depends on the system dimensions. The pressure at the auditory canal coupled through the antisymmetric mode therefore has the form shown in the dotted curve of Fig. 10.12(b) when the sound is incident from an angle $\theta = 45°$.

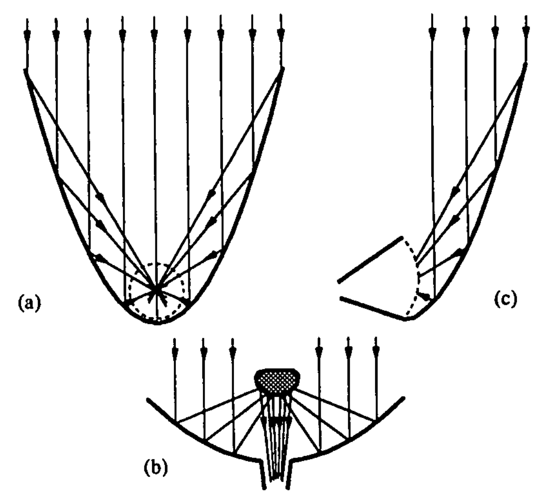
When the ear model is stimulated by a sound wave coming from along its axis, then only the plane-wave mode is excited, and the frequency response at the entry to the auditory canal is of the form shown in Fig. 10.12(a). For a sound wave incident at $\theta = 45°$, however, both modes are excited to comparable amplitudes, and their pressures add at the entry to the auditory canal, giving the frequency response shown as a full curve in Fig. 10.12(b). If the diameter of the cavity is taken to be about 20 mm, then the resonance frequency for the plane-wave mode is about 5 kHz and for the antisymmetric mode about 10 kHz, so that this mechanism may provide useful directional clues for animals such as primates whose external pinnae have something of this shape and size.

10.10 Hybrid Reflector Horns

At the limit of very high frequencies and very wide horns, say with $ka > 100$, a large number of higher modes can propagate in the horn if they are generated by unsymmetrical excitation. Instead of the analysis becoming impossibly complex in this limit, it becomes simple again, and we can think of sound rays propagating in straight lines in the same way as light, with reflection when they strike the walls of the horn. An appropriately shaped pinna may then, to some extent, focus the incident sound to the inlet meatus of the auditory system, while a badly shaped pinna, such as a deep paraboloid, may reflect it uselessly. To gain some feeling for the range of applicability of such an optical ray model, we should recall that it is impossible for an optical system to focus incident rays into an area smaller than the wavelength in diameter, and the same is true of sound. The ray approach has approximate validity, therefore, only in that part of a horn that is significantly greater than one wavelength in diameter. As the pinna narrows, it behaves first as a multi-mode and then as a single-mode horn.

Figure 10.13(a) shows the optical focusing effect achieved by a paraboloidal

Figure 10.13 (a) Focusing behavior of a paraboloidal horn when the horn diameter is much greater than the sound wavelength. The region of focus, shown dotted, is about a wavelength in diameter. (b) In the case of a very shallow primary horn, addition of a small reflector, appropriately positioned, can focus the reflected waves into an aperture. Both reflector and aperture must be larger than about one wavelength in diameter. (c) Addition of a subsidiary horn near the position of focus allows further gain. The mouth of the subsidiary horn must be about one wavelength in diameter.


(a) (b) (c)

reflector (or "parabolic" horn) when the wavelength is small compared with the horn dimensions. Sound waves incident parallel to the axis are focused to a point above the base of the paraboloid, and thence out into the free air again, so that such a horn with a small aperture in its base would not have significant gain at the high frequencies for which the optical model applies. In an optical telescope this problem would be resolved by having a secondary mirror near the focus to redirect the light to the exit port, and indeed there is a small fleshy flap in some vertebrate ears that may perhaps serve this purpose to some extent, as illustrated in Fig. 10.13(b). Such an arrangement is possible, however, only for quite shallow ears, since the reflector must be near the focus of the main paraboloid, which is at a distance from its base equal to half of the radius of a sphere that matches its curvature. The focusing action of a parabolic reflector is, however, not perfect as one would expect geometrically, for the properties of waves mean that the energy cannot be focused into a sphere smaller than about one wavelength in diameter. Both the reflecting flap and the entry to the auditory canal must therefore be at least a wavelength in diameter for such a device to work efficiently, so that its use is confined to high frequencies.

As an alternative, and more common, approach to further concentrating the sound energy in the diffraction-limited focus sphere, we can use a subsidiary horn as shown in Fig. 10.13(c). The mouth of the horn must be about one wavelength in diameter in order to collect most of the focused sound energy, and can then guide it down to a much smaller throat. The model strictly implies a roughly rectangular horn section, but clearly a circular horn would work nearly as well. In animal pinnae the transition from a reflecting to a horn-like structure is understandably less well defined than in this simple model.

The extra acoustic gain achievable with a horn-and-reflector structure such as this, above that of the horn itself, is $G_R = 10\log_{10}(S_R/S_1)$ where S_R is the projected area of the reflector in the direction of the paraboloid axis and S_1 is the mouth area of the horn. This result applies, however, only at frequencies high enough that the horn mouth is larger in diameter than the wavelength and the reflector much larger. At lower frequencies where this condition is no longer met, the gain reduces to essentially that available from the horn itself, together with the small enhancement from the pinna flap discussed in Section 10.7.

At high frequencies where focusing is efficient, the acoustic axis of the hybrid horn is parallel to the axis of the original paraboloid. At lower frequencies the shape of the reflector becomes much less important, and the acoustic axis moves away from this direction, the detailed behavior depending upon the relative sizes of horn and reflector.

Hybrid horns of this type, in which the outer part functions as a reflector rather than a horn, are perhaps the basis of the pinnae of animals such as bats that use ultrasonic frequencies. The focusing action of the outer part of the pinna can be

particularly effective if the frequency is high enough for ray-like reflection to occur, but qualitative discussion using such a reflection model at low frequencies is misleading.

References

Propagation in pipes: [1] Ch 6; [8] Ch 8; [9] Ch 5; [10] Ch 8; [11]
Horns: [1] Ch 6; [6] Ch 9; [9] Ch 5; [10] Ch 8; [25]
Short horns: [9] Ch 5; [10] Ch 8; [25]
Obliquely truncated horns: [25]
Shallow asymmetric horns: [23]; [24]

Discussion Examples

1. Calculate the resonance frequencies of a pipe of length l and cross section S, closed at one end by a slack diaphragm of thickness d.
2. Repeat question 1 for the case in which there are identical diaphragms on each end of the pipe.
3. Write a computer program to evaluate the impedance coefficients Z_{ij}, and to plot them in the form $\log(|Z_{ij}|/Z_0)$ for a cylindrical pipe and for a conical horn (neglecting wall losses). Note the infinities and zeros.
4. Write a computer program to evaluate the pressure in the throat of a horn (a conical horn is simplest) exposed to an axially incident plane wave, when the throat is rigidly stopped [equation (10.23)], and to plot this as a function of frequency. Examine the effect of changing the length of the horn, keeping mouth and throat diameters constant.
5. Write a computer program to calculate the pressure in the throat of a horn (a conical horn is simplest) exposed to an axially incident plane wave, when the throat is blocked by an arbitrary real impedance Z_L [equation (10.25)], and to plot this as a function of frequency. Find the effect of changing Z_L from a value much less than $\rho c/S_2$ to a value much greater than this quantity.

Solutions

1. The system will oscillate at its resonance frequency even after an external exciting force is removed. Write down the two network equations, neglecting radiation for simplicity, and require consistency. This gives $Z_{22}(Z_{11}+Z_T) = Z_{12}^2$. Substitute $Z_T = j\omega\rho_S d/S$ and expressions for the Z_{ij}. Simplifying gives $\tan kl = -\omega\rho_S d/S$. As $d \to 0$ resonances are $kl = n\pi$ as for an open pipe. For large d, resonances approach $kl = (n - {}^1\!/_2)\pi$ as for a pipe stopped at one end. All intermediate values are possible.
2. Proceed as in example 1. Resonance condition is $Z_{11}+Z_T = \pm Z_{12}$. As $d \to 0$, resonances are $kl = n\pi$ as for open pipe. For large d resonances approach $(n-1)\pi$ with the diaphragms moving appreciably only for the lowest resonance $n = 1$.

Horns, Strings, and Harmony

Arthur H. Benade

SCIENCE STUDY SERIES

Published by
Anchor Books
Doubleday & Company, Inc.
Garden City, New York

COVER DESIGN BY GEORGE GIUSTI
ILLUSTRATIONS BY R. PAUL LARKIN

Library of Congress Catalog Card Number 60–10663

Printed in the United States of America

Organization TC2600 Bldg/Room KNOX
United States Patent and Trademark Office
P.O. Box 1450
Alexandria, VA 22313-1450
If Undeliverable Return in Ten Days

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